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Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	**Application No.**	**Applicant(s)**	
	09/750,015	SIMARD ET AL.	
	Examiner	**Art Unit**	
	Anthony T Ton	2661	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *29 December 2000*.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-71* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-71* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☒ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on *29 December 2000* is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application (PTO-152)
6) ☐ Other: _____.

DETAILED ACTIONS

Specification

1. The disclosure is objected to because of the following informalities:

a) Term "**terminals 22, 242, 26**" in page **14** line **29** is improper. This would be a typo on the character "242".

Examiner suggests changing this term to "**terminals 22, 24, 26**".

b) Term "**jitter buffers 72**" in page **33** line **26** is not proper because it is not appropriate with Fig.9. This would be a typo on the character "72".

Examiner suggests changing this term to "**jitter buffers 122**".

Appropriate correction is required.

Claim Objections

2. **Claims 10, 32, 38, 53, 70** and **71** are objected to because of the following informalities:

a) Phrase "**media signals and output**" in **Claims 10** and **32** in line **10** is not adequate. A comma "," should be inserted between term "**media signals**" and term "**and output**" to make the phrase more clear.

Examiner suggests changing this phrase to "media signals, and output".

b) Phrase "**media signal and output**" in **Claim 38** in line 13 and **Claim 53** in line **5** is not adequate. A comma "," should be inserted between term "**media signal**" and term "**and output**" to make the phrase more clear.

Examiner suggests changing this phrase to "media signal, and output".

c) Phrase "**conference bridge and output**" in **Claims 70 and 71** in line **8** is not adequate. A comma "," should be inserted between term "**conference bridge**" and term "**and output**" to make the phrase more clear.

Examiner suggests changing this phrase to "conference bridge, and output".

Appropriate correction is required.

Claim Rejections - 35 USC § 112

3. The following is a quotation of the second paragraph of 35 U.S.C. 112:

> The specification shall conclude with one or more claims particularly pointing out and distinctly claiming the subject matter, which the applicant regards as his invention.

4. **Claims 1-13, 15-21, 23-36 and 69-71** are rejected under 35 U.S.C. 112, second paragraph, as being indefinite for failing to particularly point out and distinctly claim the subject matter which applicant regards as the invention.

a) **Claims 1-13** and **69**: **Claim 1** recites the limitation "**an output unit, coupled to the input unit**" in line **13** is vague and indefinite since there is no any specific disclosure for this limitation in the specification that relates to "**an output unit** and **an input unit**" of a conference bridge such that they are <u>coupled to each other</u>.

In addition, such a connection cannot be found in any Figures of the Drawings. However, referring to **Fig.4**, an inputting apparatus 30 (input unit) that is coupled to an outputting apparatus 34 (output unit) <u>via</u> a talker selection and mixing block 42. Therefore, this limitation of the claim it is improper.

b) **Claims 15-21** and **70**: **Claim 15** recites the limitation "**an output unit, coupled to the input unit, that operates to output addressing control signals**

to the sources" in lines **17-18** is vague and indefinite since there is no any specific disclosure for this limitation in the specification that relates to such an "**output unit**", which coupled to an input unit, operates to output "**addressing control signals**" to the sources.

In addition, such "**an output unit**" cannot be found in any Figures of the Drawings. However, referring to **Fig.7**, an energy detection and talker selection block 100, which is coupled to an inputting apparatus 30 (input unit), operates to output the "addressing controls signals". Therefore, this limitation of the claim it is improper.

c) **Claims 23-36** and **71**: **Claim 23** recites the limitation "**an output unit, coupled to the talker selection unit, that operates to output addressing control signals to the sources**" in lines **11-13** is vague and indefinite since there is no any specific disclosure for this limitation in the specification that relates to such an "**output unit**", which coupled to an input unit, operates to output "**addressing control signals**" to the sources.

In addition, such "**an output unit**" cannot be found in any Figures of the Drawings. However, referring to **Fig.11**, a talker selection block 150 that outputs the "addressing controls signals". Therefore, this limitation of the claim it is improper.

Claim Rejections - 35 USC § 102

5. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (e) the invention was described in (1) an application for patent, published under section 122(b), by another filed in the United States before the invention by the applicant for patent or (2) a patent

granted on an application for patent by another filed in the United States before the invention by the applicant for patent, except that an international application filed under the treaty defined in section 351(a) shall have the effects for purposes of this subsection of an application filed in the United States only if the international application designated the United States and was published under Article 21(2) of such treaty in the English language.

6. **Claims 1, 2, 4, 6-8, 10-14, 62 and 69** are rejected under 35 U.S.C. 102(e) as being anticipated by **Strawczynski** (US Patent No. **6,522,633**).

The applied reference has a common assignee with the instant application. Based upon the earlier effective U.S. filing date of the reference, it constitutes prior art under 35 U.S.C. 102(e). This rejection under 35 U.S.C. 102(e) might be overcome either by a showing under 37 CFR 1.132 that any invention disclosed but not claimed in the reference was derived from the inventor of this application and is thus not the invention "by another," or by an appropriate showing under 37 CFR 1.131.

a) **In Regarding to Claim 1**: **Strawczynski disclosed** a conference bridge comprising:

an input unit that operates to receive at least one media data packet from at least two sources forming a media conference, each media data packet defining a media signal *(see Fig.6 (a media conference): 538 and 540 (input unit); 521-523 (media data packets); 502-504 (at least two sources); 543-545 (media signals))*;

a talker selection unit that operates to receive speech indication signals from at least one of the sources within the media conference and to process the speech indication signals including selecting a set of the sources within the media conference as talkers *(see Fig.6: 550 and 540 including control line 556 (talker selection unit); wherein the conference algorithm circuit 550 operates to process the speech indication signals)*; and

an output unit, coupled to the input unit, that operates to output the media signals that correspond to the set of sources within the media conference selected as talkers *(see Fig.6: 552 (output unit); the connections connected from 540 to 552*

(coupling); 502-504 (a set of sources) can be selected as talkers that depends on the signal energy levels of speech signals 543-545, respectively).

b) **In Regarding to Claim 2**: **Strawczynski further disclosed** each of the speech indication signals comprises one of a talking indication and a listening indication corresponding to the respective source within the media conference *(see col.7 lines 50-60: the signal energy levels of each participant's speech are forwarded to the conference algorithm circuit 550, and it is decided which speakers are the dominant speakers (hence talking indication and listening indication)).*

c) **In Regarding to Claim 4**: **Strawczynski further disclosed** each of the speech indication signals comprises at least one speech parameter corresponding to the respective source within the media conference *(see col.9 lines 4-9: threshold energy).*

d) **In Regarding to Claim 6**: **Strawczynski further disclosed** ach of the speech indication signals is an energy level corresponding to media signals sent from the respective source within the media conference *(see col.7 lines 50-56: signal energy values of speech signals).*

e) **In Regarding to Claim 7**: **Strawczynski further disclosed** the talker selection unit operates to:

determine which sources within the media conference are sending media signals containing speech with the use of the energy levels within the speech indication signals *(see col.7 lines 56-60: it is decided which speakers are the dominant speakers)*; and

select sources within the media conference as talkers based upon the comparative energy levels of the sources within the media conference determined to be sending media signals containing speech *(see col.3 lines 25-42: the signal energy in each of the*

input communication paths to obtain a signal energy estimation, comparing each of said signal energy estimations against a predetermined threshold to obtain a signal energy comparisons).

f) **In Regarding to Claim 8**: **Strawczynski further disclosed** the speech parameter within each of the speech indication signals is a pitch value corresponding to media signals sent from the respective source within the media conference *(see col.7 lines 1-5: loudest, or "dominant" (a pitch value)).*

g) **In Regarding to Claim 10**: **Strawczynski further disclosed** the set of the sources within the media conference selected as talkers comprises a plurality of sources within the media conference *(see Fig.6: mobile participants 502-504)*; and

wherein the conference bridge further comprises a mixing block, coupled between the input and output units, that operates to receive media signals corresponding to sources within the media conference selected as talkers from the input unit, mix these received media signals and output the mixed result to the output block *(see Fig.6: Block 572, conference summing circuit).*

h) **In Regarding to Claim 11**: **Strawczynski further disclosed** the set of the sources within the media conference selected as talkers comprises a lone source within the media conference *(see col.8 line 66 – col.9 line 3: one selector input).*

i) **In Regarding to Claim 12**: **Strawczynski further disclosed** the media data packets are audio data packets and the media signals defined by the media data packets are audio signals *(see col.1 lines 15-18: audio teleconference, packets or frames; see Fig.6: PSTN participants 505-506; and see col.7 lines 50-56: PCM frames).*

j) **In Regarding to Claim 13**: **Strawczynski further disclosed** the media data packets are audio/video data packets and the media signals defined by the media data packets are audio/video signals *(see col.1 lines 11-18: The conferencing arrangement may involve voice or non-voice signals, including data, video, and facsimile signals, packets or frames).*

k) **In Regarding to Claim 14**: **Strawczynski disclosed** a conference bridge comprising:

means for receiving at least one media data packet from at least two sources forming a media conference, each media data packet defining a media signal *(see col.2 lines 1-5)*;

means for receiving speech indication signals from at least one of the sources within the media conference *(see col.2 lines 5-7)*;

means for processing the speech indication signals including selecting a set of the sources within the media conference as talkers *(see Fig.6: 550; the conference algorithm circuit 550 operates to process the speech indication signals)*; and

means for outputting the media signals that correspond to the set of sources within the media conference selected as talkers *(see Fig.6: 552 (output unit), 502-504 (the set of sources)).*

l) **In Regarding to Claim 62**: This claim is rejected for the same reasons as **Claim 1** because the apparatus in **Claim 1** can be used to practice the method steps of **Claim 62**.

m) **In Regarding to Claim 69**: **Strawczynski further disclosed** a network incorporating a conference bridge according to claim 1 and further comprising a

plurality of sources of media signals within the media conference *(see Fig.5: PSTN network)*;

wherein each of the sources within the media conference operates to output the at least one media signal to the conference bridge along with a speech indication signal corresponding to the at least one media signal *(see Fig.6: PSTN Participants (sources) 505-507, each of these sources outputted at least one media signal (output lines at selector input circuit 542) to conference bridge (summing circuit 572) and speech indication signal to conference algorithm circuit 550 via power estimation circuit 538).*

Claim Rejections - 35 USC § 103

7. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

8. **Claim 1** is rejected under 35 U.S.C. 103(a) as being unpatentable over **Su et al.** (US Patent No. **6,463,414**) in view of **Admitted Prior Art** (Fig.2 and the specification from page 3 line 23 – page 4 line 26).

Su et al. disclosed a conference bridge comprising:

an input unit that operates to receive at least one media data packet from at least two sources forming a media conference, each media data packet defining a media signal *(see Fig.2: blocks 230 and 234 (input unit), input channels 210, 214 and 218 of*

the conference bridge 200 receiving media data packet from participants 1-3 via packet network 201); and

an output unit, coupled to the input unit, that operates to output the media signals that correspond to the set of sources within the media conference selected as talkers *(see Fig.2: blocks 232 and 234 (output unit); output channels 212, 216 and 220 outputted media signals from the conference bridge 220 to participants 1-3 via packet network 201; and see col.2 line 64- col.3 line 12: signaling process, data transmission, signaling, packet-based transmission, and network control).*

Su et al. failed to disclose a talker selection unit that operates to receive speech indication signals from at least one of the sources within the media conference and to process the speech indication signals including selecting a set of the sources within the media conference as talkers.

The Admitted Prior Art clearly disclosed such a conference bridge *(see Figure 2 of Admitted Prior Art: block 32, voice data packets; and see the specification of the Admitted Prior from page 3 line 23–page 4 line 26: conferencing algorithm).*

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify such a talker selection unit of the Admitted Prior Art to the packet-based conference bridge as taught by Su et al. in order to select talkers in the conference for appropriate connections in a conferencing system, **the motivation being** to make operate all participants in a conferencing system more efficient.

9. **Claims 2-5, 9, 23-36** and **71** are rejected under 35 U.S.C. 103(a) as being unpatentable over **Su et al.** (US Patent No. **6,463,414**) in view of **Admitted Prior**

Art (Fig.2 and the specification from page 3 line 23 – page 4 line 26) as applied to claim 1 above, and further in view of **Foster et al.** (US Patent No. **6,466,550**).

a) **In Regarding to Claim 2**: **Su et al.** and **Admitted Prior Art** disclosed all aspects of claim 2 as set forth in claim 1.

Su et al. failed to explicitly disclose each of the speech indication signals comprises one of a talking indication and a listening indication corresponding to the respective source within the media conference.

Forster et al. disclosed such each of the speech indication signals *(see Forster et al. Fig.6: Speaking and Heard).*

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify such each of the speech indication signals of Foster et al. to the conference bridge, as taught by Su et al. in order to detect talkers and listeners in a conference communications system properly, **the motivation being** to make Su et al. more efficient.

b) **In Regarding to Claim 3**: **Su et al. further disclosed** the talker selection unit operates to:

monitor the speech indication signals for talking indications *(see col.7 lines 39-53: speech parameters are monitored)*; and

select sources within the media conference as talkers based upon the order in which any talking indications are received at the talker selection unit from the sources within the media conference *(see Fig.5: block 560 priorities assignment).*

It would have been obvious to combine **Su et al.**, **Forster et al.** and the **Admitted Prior Art** for the same reason as in claims 1 and 2.

c) **In Regarding to Claim 4**: **Su et al. further disclosed** each of the speech indication signals comprises at least one speech parameter corresponding to the respective source within the media conference *(see col.2 lines 15-21: incoming speech channels associated with multiple participants).*

It would have been obvious to combine **Su et al.**, **Forster et al**. and the **Admitted Prior Art** for the same reason as in claim 1.

d) **In Regarding to Claim 5**: **Su et al further disclosed** that the talker selection unit operates to:

determine which sources within the media conference are sending media signals containing speech with the use of the speech parameters within the speech indication signals *(see col.7 lines 44-53: speech parameters are monitored to determine which speaker is in fact dominating the discussion (hence, determine which sources within the media conference are sending media signals containing speech)*); and

select sources within the media conference as talkers based upon the order in which sources within the media conference are determined to send media signals containing speech *(see Fig.5: block 560 priorities assignment).*

It would have been obvious to combine **Su et al.**, **Forster et al**. and the **Admitted Prior Art** for the same reason as in claims 1 and 4.

e) **In Regarding to Claim 9**: **Su et al. further disclosed** each of the speech indication signals is a number of bytes within media signals sent from the respective source within the media conference *(see col.4 lines 8-19: speech coding/compression techniques; and see G.723.1; the G.723.1 is a VoIP standard well known in the art (hence, number of bytes in a voice data packet)).*

It would have been obvious to combine **Su et al.**, **Forster et al**. and the **Admitted Prior Art** for the same reason as in claims 1 and 4.

f) **In Regarding to Claim 23**: **Su et al. disclosed** a conference bridge arranged to be coupled to packet-based network that including at least two sources of media signals forming a media conference *(see Fig.2)*, the conference bridge comprising:

an output unit, coupled to the input unit *(see Fig.5: blocks 515 and 550)*, that operates to output addressing control signals to the sources within the media conference selected as talkers *(see col.2 line 64- col.3 line 12: signaling process, data transmission, signaling, packet-based transmission, and network control (hence addressing control signals))*.

Su et al. failed to disclose a talker selection unit that operates to receive speech indication signals from at least one of the sources within the media conference and to process the speech indication signals including selecting a set of the sources within the media conference as talkers.

The Admitted Prior Art clearly disclosed such a conference bridge *(see Figure 2 of Admitted Prior Art: block 32, voice data packets; and see the specification of the Admitted Prior from page 3 line 23–page 4 line 26: conferencing algorithm)*.

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify such a talker selection unit of the Admitted Prior Art to the packet-based conference bridge as taught by Su et al. in order to select talkers in the conference for appropriate connections in a conferencing system, **the motivation being** to make operate all participants in a conferencing system more efficient.

Su et al. also failed to explicitly disclose the addressing control signals comprising instructions for the sources within the media conference selected as talkers to output their media signals directly to other sources within the media conference. **Su et al. implied** teaching a direct communication in a conference involving three or more participants in connection with person-to-person transcoding (*see col. 5 lines 32-38*). In addition, Su et al. mentioned such a direct communication between two participants (*see col.3 lines 15-33*). **Therefore**, it would be obvious for such a direction communication between participants in a conferencing system as taught by Su et al.

However, Foster et al. clearly disclosed such addressing control signals (*see col.1 line 64–col.2 line 13: using multicast packet transmission (participants communicate directly to each other); and see col.6 lines 8-20: Real-time Transport Protocol (hence, instructions)).*

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify such instructions of Foster et al. to the voice over a packet-based network as taught by Su et al. in order to provide digitally encoded voice that can be transmitted directly over a packet-based network to participants in a conference communications system, **the motivation being** to make Su et al. more efficient.

g) **In Regarding to Claim 24**: **Su et al.** and **Admitted Prior Art** disclosed all aspects of claim 24 as set forth in claim 23.

Su et al. failed to explicitly disclose each of the speech indication signals comprises one of a talking indication and a listening indication corresponding to the respective source within the media conference.

Forster et al. disclosed such each of the speech indication signals *(see Forster et al. Fig.6: Speaking and Heard).*

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify such each of the speech indication signals of Foster et al. to the conference bridge, as taught by Su et al. in order to detect talkers and listeners in a conference communications system properly, **the motivation being** to make Su et al. more efficient.

h) **In Regarding to Claim 25**: **Su et al. further disclosed** the talker selection unit operates to:

monitor the speech indication signals for talking indications *(see col.7 lines 39-53: speech parameters are monitored)*; and

select sources within the media conference as talkers based upon the order in which any talking indications are received at the talker selection unit from the sources within the media conference *(see Fig.5: block 560 priorities assignment).*

It would have been obvious to combine **Su et al.**, **Forster et al**. and the **Admitted Prior Art** for the same reason as in claim 23.

i) **In Regarding to Claim 26**: **Su et al. further disclosed** each of the speech indication signals comprises at least one speech parameter corresponding to the respective source within the media conference *(see col.2 lines 15-21: incoming speech channels associated with multiple participants).*

It would have been obvious to combine **Su et al.**, **Forster et al**. and the **Admitted Prior Art** for the same reason as in claim 23.

j) **In Regarding to Claim 27**: **Su et al further disclosed** that the talker selection unit operates to:

determine which sources within the media conference are sending media signals containing speech with the use of the speech parameters within the speech indication signals *(see col.7 lines 44-53: speech parameters are monitored to determine which speaker is in fact dominating the discussion (hence, determine which sources within the media conference are sending media signals containing speech*)); and

select sources within the media conference as talkers based upon the order in which sources within the media conference are determined to send media signals containing speech *(see Fig.5: block 560 priorities assignment).*

It would have been obvious to combine **Su et al.**, **Forster et al.** and the **Admitted Prior Art** for the same reason as in claim 23.

k) **In Regarding to Claim 28**: **Su et al. further disclosed** the speech parameter within each of the speech indication signals is an energy level corresponding to media signals sent from the respective source within the media conference *(see col.6 lines 11-30: a set of parameters, e.g., spectrum, pitch, energy).*

It would have been obvious to combine **Su et al.**, **Forster et al.** and the **Admitted Prior Art** for the same reason as in claim 23.

l) **In Regarding to Claim 29**: **Su et al. further disclosed** the talker selection unit operates to:

determine which sources within the media conference are sending media signals containing speech with the use of the energy levels within the speech indication signals *(see col.7 lines 44-53: speech parameters are monitored to determine which speaker is*

in fact dominating the discussion (hence, determine which sources within the media conference are sending media signals containing speech)); and

select sources within the media conference as talkers based upon the comparative energy levels of the sources within the media conference determined to be sending media signals containing speech *(see Fig.5: block 560 priorities assignment).*

It would have been obvious to combine **Su et al.**, **Forster et al.** and the **Admitted Prior Art** for the same reason as in claim 23.

m) **In Regarding to Claim 30**: **Su et al. further disclosed** the speech parameter within each of the speech indication signals is a pitch value corresponding to media signals sent from the respective source within the media conference *(see col.6 lines 11-30: a set of parameters, e.g., spectrum, pitch; and see col.7 lines 31-37: pitch parameters).*

It would have been obvious to combine **Su et al.**, **Forster et al.** and the **Admitted Prior Art** for the same reason as in claim 23.

n) **In Regarding to Claim 31**: **Su et al. further disclosed** each of the speech indication signals is a number of bytes within media signals sent from the respective source within the media conference *(see col.4 lines 8-19: speech coding/compression techniques; and see G.723.1; the G.723.1 is a VoIP standard well known in the art (hence, number of bytes in a voice data packet)).*

It would have been obvious to combine **Su et al.**, **Forster et al.** and the **Admitted Prior Art** for the same reason as in claim 23.

o) **In Regarding to Claim 32**: **Su et al. further disclosed** the set of the sources within the media conference selected as talkers comprises a plurality of sources within the media conference *(see Fig.2: participants 1-3)*; and

wherein the conference bridge further comprises a mixing block, coupled between the input and output units, that operates to receive media signals corresponding to sources within the media conference selected as talkers from the input unit, mix these received media signals and output the mixed result to the output block *(see Fig.2: blocks 238 (mixer), 230 (input), and 232 (output), input channels 210, 214 and 218 that operate to receive media signals from sources 1-3, respectively; and output channels that operate to output media signals from mixers 1-3 of the conference bridge 200 to sources 1-3, respectively)*.

It would have been obvious to combine **Su et al.**, **Forster et al**. and the **Admitted Prior Art** for the same reason as in claim 23.

p) **In Regarding to Claim 33**: **Su et al. further disclosed** the set of the sources within the media conference selected as talkers comprises a lone source within the media conference *(see col.7 lines 54-60: a single participant)*.

It would have been obvious to combine **Su et al.**, **Forster et al**. and the **Admitted Prior Art** for the same reason as in claim 23.

q) **In Regarding to Claim 34**: **Su et al. further disclosed** the addressing control signals comprise packet-based network addresses corresponding to the other sources within the media conference *(see col.2 line 64- col.3 line 12: signaling process, data transmission, signaling, packet-based transmission, and network control (hence addressing control signals))*.

It would have been obvious to combine **Su et al.**, **Forster et al**. and the **Admitted Prior Art** for the same reason as in claim 23.

r) **In Regarding to Claim 35**: **Su et al. further disclosed** the media data packets are audio data packets and the media signals defined by the media data packets are audio signals *(see col.4 lines 8-17: at least one packet-based voice channel with a number of other voice channels).*

It would have been obvious to combine **Su et al.**, **Forster et al**. and the **Admitted Prior Art** for the same reason as in claim 23.

s) **In Regarding to Claim 36**: **Su et al.** and **Admitted Prior Art** disclosed all aspects of claim 36 as set forth in claim 23

However, **Su et al. failed to explicitly disclose** the media data packets are audio/video data packets and the media signals defined by the media data packets are audio/video signals.

Foster et al. disclosed such media packets *(see col.11 lines 20-31: both voice and video packets).*

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify such audio/video data packets of Foster et al. to the voice over a packet-based network as taught by Su et al. in order to provide a voice-controlled video conference over a packet-based network to participants in a conferencing system, **the motivation being** to make Su et al. applicable in a video conference.

t) **In Regarding to Claim 71**: **Su et al further disclosed** a network incorporating a conference bridge according to claim 23 and further comprising a

plurality of sources of media signals within the media conference *(see Figs. 1 and 2: packet network)*;

wherein each of the sources within the media conference operates to output a speech indication signal to the conference bridge, receive the addressing control signal from the conference bridge and output their media signals to the other sources within the media conference based upon the received addressing control signal *(see Fig.2: participants 1-3, conference bridge 200; and see col.5 line 19-col.6 line 30: in a conference involving three or more participants, invention may also be employed in connection with person-to-person transcoding (hence this connection would be considered by the conference bridge); also, see col.1 lines 11-14: VoIP (hence including audio and addressing control signals))*.

It would have been obvious to combine **Su et al.**, **Forster et al**. and the **Admitted Prior Art** for the same reason as in claim 23.

10. **Claim 15** is rejected under 35 U.S.C. 103(a) as being unpatentable over the **Admitted Prior Art (Fig.2** and **the specification page 3 line 23 – page 4 line 26)** in view of **Alvarez, III et al. (**US Patent No. **4,507,781)**.

Based on the disclosures in Fig.2 and the specification (from page 3 line 23 to page 4 line 26) of the Admitted prior art, most of subject matters of the claimed limitation of this claim **were disclosed by the Admitted prior art**.

Only one difference between the **Admitted Prior Art** and the instant claim is that an output unit that operates to output addressing control signals to sources within a media conference selected as talkers, the addressing control signals comprising

instructions for the sources within the media conference selected as talkers to output their media signals directly to other sources within the media conference.

Alvarez, III et al. clearly disclosed such the addressing control signals *(see abstract: The disclosed apparatus appends a direct destination address to each point-to-point port communication for transmission over a communication link, to directly address the intended destination port (hence, sources communicate directly to one another); and see col.64 line 24 – col.65 line 2: address and control signal information).*

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify such the addressing control signals and point-to-point and multipoint of Alvarez et al. to the voice over a packet-based network in a media conference in order to provide digitally encoded voice that can be transmitted directly over a packet-based network to participants in a conference communications system, **the motivation being** to make a media conference more efficient.

11. **Claims 15-21** and **70** are rejected under 35 U.S.C. 103(a) as being unpatentable over **Su et al.** (US Patent No. **6,463,414**) in view of **Foster et al.** (US Patent No. **6,466,550**).

a) **In Regarding to Claim 15**: **Su et al. disclosed** a conference bridge comprising:

an input unit that operates to receive at least one media data packet from at least two sources forming a media conference, each media data packet defining a media signal *(see Fig.2: the combination of blocks 230, 234 and 238-242 is considered as an*

input unit that operates to receive at least one media data packet from at least two sources forming a media conference; and see col.1 lines 11-14 (voice over packet networks (hence, each media data packet defining a media signal)).

an energy detection and talker selection unit, coupled to the input unit *(see col.4 lines 28-34: an intelligent scheme (energy and talker selection unit); and see Fig.3: blocks 302 and 308)*, that operates to:

determine at least one speech parameter corresponding to each of the media signals *(see col.6 lines 1-15: speech parameters)*; and

select a set of the sources within the media conference as talkers based on the determined speech parameters *(see col.4 lines 28-41: parameter extraction (select a set of the source); and see col.7 lines 40-53: priority level)*; and

an output unit, coupled to the input unit *(see Fig.5: blocks 515 and 550)*, that operates to output addressing control signals to the sources within the media conference selected as talkers *(see col.2 line 64- col.3 line 12: signaling process, data transmission, signaling, packet-based transmission, and network control (hence addressing control signals))*.

Su et al. failed to explicitly disclose the addressing control signals comprising instructions for the sources within the media conference selected as talkers to output their media signals directly to other sources within the media conference. **Su et al. implied** teaching a direct communication in a conference involving three or more participants in connection with person-to-person transcoding (*see col. 5 lines 32-38*).

In addition, Su et al mentioned such a direct communication between two participants *(see col.3 lines 15-33)*. **Therefore**, it would be obvious for such a direction communication between participants in a conferencing system as taught by Su et al.

However, Foster et al. clearly disclosed such addressing control signals *(see col.1 line 6–col.2 line 13: using multicast packet transmission (participants communicate directly to each other); and see col.6 lines 8-20: Real-time Transport Protocol (hence, instructions)).*

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify such instructions of Foster et al. to the voice over a packet-based network as taught by Su et al. in order to provide digitally encoded voice that can be transmitted directly over a packet-based network to participants in a conference communications system, **the motivation being** to make Su et al. more efficient.

b) **In Regarding to Claim 16**: **Su et al further disclosed** the addressing control signals comprise packet-based network addresses corresponding to the other sources within the media conference *(see col.2 line 64- col.3 line 12: signaling and network control (hence addressing control signals)).*

It would have been obvious to combine **Su et al.** and **Forster et al**. for the same reason as in claim 15.

c) **In Regarding to Claim 17**: **Su et al further disclosed** the media data packets are audio data packets and the media signals defined by the media data packets are compressed audio signals *(see col.1 lines 18-22: VoIP)*; and

wherein the speech parameter corresponding to each of the media signals is a number of bytes within each of the compressed audio signals *(see col.4 lines 8-17: speech coding/compress techniques (compress), the specification of G.723.1 VoIP standard (hence, number of bytes in a voice data packet)).*

It would have been obvious to combine **Su et al.** and **Forster et al**. for the same reason as in claim 15.

d) **In Regarding to Claim 18**: **Su et al further disclosed** the speech parameter corresponding to each of the media signals is a pitch value corresponding to each of the media signals *(see col.7 line 66-col.8 line 3: pitch parameter).*

It would have been obvious to combine **Su et al.** and **Forster et al**. for the same reason as in claim 15.

e) **In Regarding to Claim 19**: **Su et al further disclosed** the speech parameter corresponding to each of the media signals is an energy level corresponding to each of the media signals *(see col.7 lines 25-37: the signal driven e.g., based on the relative energy of the talker).*

It would have been obvious to combine **Su et al.** and **Forster et al**. for the same reason as in claim 15.

f) **In Regarding to Claim 20**: **Su et al further disclosed** the media data packets are audio data packets and the media signals defined by the media data packets are audio signals *(see col.4 lines 8-17: packet-based voice channel).*

It would have been obvious to combine **Su et al.** and **Forster et al**. for the same reason as in claim 15.

g) **In Regarding to Claim 21**: **Su et al. disclosed** all aspects of the claim 21 as set forth in claim 15.

However, **Su et al. failed to explicitly disclose** the media data packets are audio/video data packets and the media signals defined by the media data packets are audio/video signals.

Foster et al. disclosed such media packets *(see col.11 lines 20-31: both voice and video packets).*

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify such audio/video data packets of Foster et al. to the voice over a packet-based network as taught by Su et al. in order to provide a voice-controlled video conference over a packet-based network to participants in a conferencing system, **the motivation being** to make Su et al. applicable in a video conference.

h) **In Regarding to Claim 70**: **Su et al further disclosed** a network incorporating a conference bridge according to claim 15 and further comprising a plurality of sources of media signals within the media conference *(see Figs.1 and 2: packet network)*;

wherein each of the sources within the media conference operates to output the at least one media signal to the conference bridge, receive the addressing control signal from the conference bridge and output their media signals to the other sources within the media conference based upon the received addressing control signal *(see Fig.2: participants 1-3, conference bridge 200; and see col.5 line 19-col.6 line 30: in a conference involving three or more participants, invention may also be employed in connection with person-to-person transcoding (hence this connection would be*

considered by the conference bridge); also, see col.1 lines 11-14: VoIP (hence including audio and addressing control signals)) .

It would have been obvious to combine **Suet al.** and **Forster et al.** for the same reason as in claim 15.

12. **Claims 38 and 63** are rejected under 35 U.S.C. 103(a) as being unpatentable over **the Admitted Prior Art (Fig.5** and Applicant' specification from page 20 line 25 to page 21 line 2) in view of **Strawczynski et al.** (US Patent No. **4,920,565**).

a) **In Regarding to Claim 38**: **The Admitted Prior Art disclosed** a packet-based apparatus arranged to be coupled to a conference bridge via a packet-based network, the packet based apparatus comprising:

an output unit that operates to receive at least one media signal from at least one participant within a media conference and output the received media signal to the conference bridge via the packet-based network *(see Applicant' specification page 20 line 25 – page 21 line 2, and see Fig.5: blocks 60, 62 and 64 (output unit). All of blocks depicted in Fig.5 exclusive block 66 are well known in the art).*

The Admitted Prior Art failed to explicitly disclose a speech detection unit, coupled to the output unit, that operates to process the received media signal, generate a speech indication signal based upon the received media signal and output the speech indication signal to the conference bridge. However, **Strawczynski et al. clearly disclosed** such a speech detection unit *(see Fig.4: 44 (speech detector); 43, 60*

and 61 (generate a speech indication signal); and 62 (outputted speech indication signal to the conference bridge via a network)).

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify the packet-based apparatus of the Applicant by utilizing the speech detector 44, as taught by Strawczynski et al., so that a detected signal can be forwarded to a conference bridge appropriately, **the motivation being** to reduce the time of processing at the conference bridge thus enhancing efficiency.

b) **In Regarding to Claim 63**: This claim is rejected for the same reasons as **Claim 38** because the apparatus in **Claim 38** can be used to practice the method steps of **Claim 63**.

13. **Claims 22, 37-40, 63 and 64** are rejected under 35 U.S.C. 103(a) as being unpatentable over **Strawczynski et al.** (US Patent No. **4,920,565**) in view of **DuVal** (US Patent No. **5,818,836**).

a) **In Regarding to Claim 22**: **Strawczynski et al. disclosed** a conference bridge comprising:

means for receiving at least one media data packet from at least two sources forming a media conference, each media data packet defining a media signal *(see Fig.6: DEMUX, ch.1 and ch.2)*;

means for selecting a set of the sources within the media conference as talkers *(see col.6 lines 16-25: conference control unit 80)*; and

means for instructing the sources within the media conference selected as talkers to output their media signals directly to other sources within the media conference *(see*

<u>Figs. 1 and 2 and col.3 lines 41-46</u>: conference control unit (means for instructing); based on the connections established in Figs. 1 and 2, the telephones would be connected directly to each other and the conference control unit 11 acts to set up and supervise the interconnection between the telephones to form the conference. No processing of speech by the conference unit or by the network is required).

Strawczynski et al. failed to explicitly disclose the network that is a packet-based network. **However, Strawczynski et al**. disclosed that telephone sets in the most practical arrangement, digital communication techniques are used for access, transmission and switching. A method can still be applied to an analog or a mixed analog/digital switching/transmission network if suitable low bit rate speech coding and voice band data modems are employed to provide digital signals for use by the secure telephone sets *(<u>see col.3 lines 29-40</u>)*. **Therefore**, it would be obvious for any packet-based terminals, which operate in a packet-based network, can be communicated to one another if using voice band data modems, as taught by Strawczynski et al. **DuVal clearly disclosed** such a packet-based network *(<u>see Fig.1</u>: block 16)*.

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify such a packet-based network of DuVal to the network as taught by Strawczynski et al. in order to provide an anonymous telephone communication system over a packet-based network, **the motivation being** to utilize digitally encoded voice as well as data over packet-based networks.

b) **In Regarding to Claim 37**: **Strawczynski et al. disclosed** a conference bridge arranged to be coupled to a network that includes at least two sources of media signals forming a media conference *(see Fig.6)*, the conference bridge comprising:

means for receiving speech indication signals from at least one of the sources within the media conference *(see Fig.6: DEMUX, ch.1 and ch.2)*;

means for processing the speech indication signals including selecting a set of the sources within the media conference as talkers *(see col.6 lines 16-25: conference control unit 80)*; and

means for instructing the sources within the media conference selected as talkers to output their media signals directly to other sources within the media conference *(see Figs. 1 and 2 and col.3 lines 41-46: conference control unit (means for instructing); based on the connections established in Figs. 1 and 2, the telephones would be connected directly to each other and the conference control unit 11 acts to set up and supervise the interconnection between the telephones to form the conference. No processing of speech by the conference unit or by the network is required).*

Strawczynski et al. failed to explicitly disclose the network that is a packet-based network. **However, Strawczynski et al**. disclosed that telephone sets in the most practical arrangement, digital communication techniques are used for access, transmission and switching. A method can still be applied to an analog or a mixed analog/digital switching/transmission network if suitable low bit rate speech coding and voice band data modems are employed to provide digital signals for use by the secure telephone sets *(see col.3 lines 29-40)*. Therefore, it would be obvious for any packet-based terminals, which operate in a packet-based network, can be

communicated to one another if using voice band data modems, as taught by Strawczynski et al. **DuVal clearly disclosed** such a packet-based network *(see Fig.1: block 16).*

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify such a packet-based network of DuVal to the network as taught by Strawczynski et al. in order to provide an anonymous telephone communication system over a packet-based network, **the motivation being** to utilize digitally encoded voice as well as data over packet-based networks.

c) **In Regarding to Claim 38**: **Strawczynski et al. disclosed** an apparatus arranged to be coupled to a conference bridge via a network, the apparatus comprising:

an output unit that operates to receive at least one media signal from at least one participant within a media conference and output the received media signal to the conference bridge via the network *(see Fig.4: a combination of blocks 44, 41, 57, 58 and 59 is an output unit; 45, 49 and 52 (received media signal); 56 and 49 (at least one participant); 59 and 62 (output the media signal to the conference bridge via the network; and see Fig.1: 11 (conference bridge), 10 (network));* and

a speech detection unit, coupled to the output unit, that operates to process the received media signal, generate a speech indication signal based upon the received media signal and output the speech indication signal to the conference bridge *(see Fig.4: speech detector inside block 44, and block 44 is also used to process the received media (audio) signal; 43, 60 and 61 (generate a speech indication signal); and 62 (outputted the speech indication signal to the conference bridge); and see Fig.1: the connections shown in Fig.1 would indicate such an outputted indication signal).*

Strawczynski et al. failed to explicitly disclose the network that is a packet-based network. **However, Strawczynski et al. disclosed** that telephone sets in the most practical arrangement, digital communication techniques are used for access, transmission and switching. A method can still be applied to an analog or a mixed analog/digital switching/transmission network if suitable low bit rate speech coding and voice band data modems are employed to provide digital signals for use by the secure telephone sets *(<u>see col.3 lines 29-40</u>)*. Therefore, it would be obvious for any packet-based terminals, which operate in a packet-based network, can be communicated to one another if using voice band data modems, as taught by Strawczynski et al.

DuVal clearly disclosed such a packet-based network *(<u>see Fig.1</u>: block 16)*.

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify such a packet-based network of DuVal to the network as taught by Strawczynski et al. in order to provide an anonymous or analog telephone communication system over a packet-based network, **the motivation being** to utilize digitally encoded voice as well as data over packet-based networks.

d) **In Regarding to Claim 39**: **Strawczynski et al. further disclosed** the output unit comprises a microphone that operates to receive audio signals from the at least one participant within the media conference, the received media signal comprising audio signals received from the microphone *(<u>see Fig.4</u>: 56 (microphone), 45 (audio signal))*.

It would have been obvious to combine **Strawczynski et al.** and **DuVal** for the same reason as in **Claim 38**.

e) **In Regarding to Claim 40**: **Strawczynski et al. further disclosed** a network interface comprising an apparatus according to claim 38, wherein the output unit receives the media signal from the at least one participant within the media conference from a non-packet-based telephone terminal via the non-packet-based apparatus *(see Fig.4: blocks 44, 41, 51 and 50 (output unit); blocks 49 and 52 (received media signal); Fig.4 is considered as a non-packet-based apparatus; and see Fig.6: telephones A and B (non-packet-based terminals) receive the media signal from each other via Ch2 and Ch1).*

Strawczynski et al. failed to explicitly disclose a packet-based interface arranged to be coupled between a packet-based network and a non-packet-based network.

DuVal disclosed such a packet-based interface *(see Fig.2: modems 40 and 38).*

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify such a packet-based network of DuVal to the network as taught by Strawczynski et al. in order to provide an anonymous telephone or analog communication system over a packet-based network and vice versa, **the motivation being** to utilize digitally encoded voice as well as data over packet-based networks.

f) **In Regarding to Claim 63**: This claim is rejected for the same reasons as **Claim 38** because the apparatus in **Claim 38** can be used to practice the method steps of **Claim 63**.

g) **In Regarding to Claim 64**: This claim is rejected for the same reasons as **Claim 22** because the apparatus in **Claim 22** can be used to practice the method steps of **Claim 64**.

14. **Claims 38-41, 49-54, 63, 67 and 68** are rejected under 35 U.S.C. 103(a) as being unpatentable over **Dunn et al.** (US Patent No. **5,991,385**) in view of **DuVal** (US Patent No. **5,818,836**).

a) **In Regarding to Claim 38**: **Dunn et al. disclosed** an apparatus arranged to be coupled to a conference bridge via a network, the apparatus comprising:

an output unit that operates to receive at least one media signal from at least one participant within a media conference and output the received media signal to the conference bridge via the network *(see col.5 lines 13-30: the combination of blocks DAA circuit 33, Transceiver 34 and DSP 36 is an output unit, conference signals (at least one media signal); and see Fig 5: Boxes 10 and 12 (the network));* and

a speech detection unit, coupled to the output unit, that operates to process the received media signal, generate a speech indication signal based upon the received media signal and output the speech indication signal to the conference bridge *(see Fig.3: Box 36 (speech detection unit), DAA 33 and transceiver 34 (output unit); and see col.5 lines 20-30: the DSP 36).*

Dunn et al. failed to explicitly disclose the network that is a packet-based network.

DuVal clearly disclosed such a packet-based network *(see Fig.1: block 16).*

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify such a packet-based network of DuVal to the network as taught by Dunn et al. in order to provide an anonymous or analog telephone

communication system over a packet-based network, **the motivation being** to utilize digitally encoded voice as well as data over packet-based networks.

b) **In Regarding to Claim 39**: **Dunn et al. further disclosed** the output unit comprises a microphone that operates to receive audio signals from the at least one participant within the media conference, the received media signal comprising audio signals received from the microphone *(see Fig.2: 16'; and see Fig.3: 22)*

It would have been obvious to combine **Dunn et al.** and **DuVal** for the same reason as in claim 38.

c) **In Regarding to Claim 40**: **Dunn et al. disclosed** all aspects of the claim 40 as set forth in the claim 38.

Dunn et al. failed to explicitly disclose a packet-based interface arranged to be coupled between a packet-based network and a non-packet-based network.

DuVal disclosed such a packet-based interface *(see Fig.2: modems 40 and 38).*

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify such a packet-based network of DuVal to the network as taught by Dunn et al. in order to provide an anonymous telephone or analog communication system over a packet-based network and vice versa, **the motivation being** to operate both voice and data over both packet-based and non-packet-based networks in a conference.

d) **In Regarding to Claim 41**: **Dunn et al. further disclosed** the speech detection unit *(see DSP in Fig.3)* operates to:

determine if the received media signal contains speech *(col.5 lines 51-56: DSP code measures individual inputs, and the measurement of volume and power density*

or different frequencies of the inputs, this can be considered as that the DSP would determine whether the received media signal contains speech or does not contain speech);

if the received media signal contains speech, include a talking indication within the speech indication signal *(see col.5 lines 20-30: re-create the audio and output array (talking indication)); and*

if the received media signal does not contain speech, include a listening indication within the speech indication signal *(see col.6 lines 18-43: in which, DSP processing power in the speakerphone of each party creates a sound field effect for each party, and the ID signal is sent by the speakerphone is sufficient to generate a virtual table location of each participant and the outgoing image (this can considered as a listening indication) enables each speaker in a group to be identified).*

It would have been obvious to combine **Dunn et al.** and **DuVal** for the same reason as in claim 38.

e) **In Regarding to Claim 63**: This claim is rejected for the same reasons as **Claim 38** because the apparatus in **Claim 38** can be used to practice the method steps of **Claim 63**.

f) **In Regarding to Claims 49** and **50**: **Dunn et al. disclosed** an apparatus arranged to be coupled to a conference bridge via a network *(see Fig.5)*, the apparatus comprising:

an addressing control unit that operates to receive at least one addressing control signal from the conference bridge *(see Figs.2 and 3: the signaling path 18 shown in Fig.2 is used to connect a control signal from the conference 14 to Party D; Box 33 in*

Fig.3 is considered as an addressing control unit; and *see col.5 lines 13-30: conference signals that include both audio and* *port identity signals* *(addressing control signal));* and

an output unit that operates to receive at least one media signal from at least one participant within a media conference and output the received media signal, via the network, to at least one other participant within the media conference based upon the addressing control signal *(see col.5 lines 13-30: the combination of blocks DAA circuit 33, Transceiver 34 and DSP 36 is an output unit, conference signals (at least one media signal); and see Fig 5: Boxes 10 and 12 (the network)).*

Dunn et al. failed to explicitly disclose the network that is a packet-based network (**claim 49**) and addressing control signal comprises a packet-based network address (**claim 50**).

DuVal clearly disclosed such a packet-based network and such a packet-based network address *(see Fig.1: block 16 (packet-based network) and see col.11 lines 37-40: Online Data Service ID field (*packet-based network address*)).*

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify such a packet-based network of DuVal to the network as taught by Dunn et al. in order to provide an anonymous or analog telephone communication system over a packet-based network (VoIP), **the motivation being** to utilize both digitally encoded voice and data over packet-based networks.

g) **In Regarding to Claim 51**: **Dunn et al. further disclosed** the output unit comprises a microphone that operates to receive audio signals from the at least one

participant within the media conference, the received media signal comprising audio signals received from the microphone *(see Fig.2: 16'; and see Fig.3: 22)*

It would have been obvious to combine **Dunn et al.** and **DuVal** for the same reason as in claim 49.

h) **In Regarding to Claim 52**: **Dunn et al. disclosed** all aspects of the claim 52 as set forth in the claim 49.

Dunn et al. failed to explicitly disclose a packet-based interface arranged to be coupled between a packet-based network and a non-packet-based network. **DuVal disclosed** such a packet-based interface *(see Fig.2: modems 40 and 38)*.

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify such a packet-based network of DuVal to the network as taught by Dunn et al. in order to provide an anonymous telephone or analog communication system over a packet-based network and vice versa, **the motivation being** to operate both voice and data over both packet-based and non-packet-based networks in a conference.

i) **In Regarding to Claim 53**: **Dunn et al. further disclosed** an apparatus according to claim 49 further comprising a speech detection unit, coupled to the output unit, that operates to process the received media signal, generate a speech indication signal based upon the received media signal and output the speech indication signal to the conference bridge *(see Fig.3: Box 36 (speech detection unit), DAA 33 and transceiver 34 (output unit); and see col.5 lines 20-30: the DSP 36)*.

It would have been obvious to combine **Dunn et al.** and **DuVal** for the same reason as in claim 49.

j) **In Regarding to Claim 54**: **Dunn et al. further disclosed** the speech detection unit *(see DSP in Fig.3)* operates to:

determine if the received media signal contains speech *(col.5 lines 51-56: DSP code measures individual inputs, and the measurement of volume and power density or different frequencies of the inputs, this can be considered as that the DSP would determine whether the received media signal contains speech or does not contain speech)*;

if the received media signal contains speech, include a talking indication within the speech indication signal *(see col.5 lines 20-30: re-create the audio and output array (talking indication));* and

if the received media signal does not contain speech, include a listening indication within the speech indication signal *(see col.6 lines 18-43: in which, DSP processing power in the speakerphone of each party creates a sound field effect for each party, and the ID signal is sent by the speakerphone is sufficient to generate a virtual table location of each participant and the outgoing image (this can considered as a listening indication) enables each speaker in a group to be identified).*

It would have been obvious to combine **Dunn et al.** and **DuVal** for the same reason as in claim 49.

k) **In Regarding to Claim 67**: **Dunn et al. disclosed** a method for an apparatus to operate within a media conference controlled by a conference bridge, the method comprising:

receiving at least one media signal from at least one participant within the media conference *(see col.3 lines 23-44: the bridge linking together all participants, the port identity/audio signals (hence, receiving at least one media signals))* ;

receiving at least one addressing control signal from the conference bridge *(see Figs.2 and 3: the signaling path 18 shown in Fig.2 is used to connect a control signal from the conference 14 to Party D; Box 33 in Fig.3 is considered as an addressing control unit; and see col.5 lines 13-30: conference signals that include both audio and port identity signals (addressing control signal))*; and

outputting the received media signal to at least one other participant within the media conference based upon the addressing control signal *(see col.5 lines 13-30: Transceiver 34 transmits conference signals i.e. audio signal (media signal) and port identity signal (addressing control signal)).*

Dunn et al. failed to explicitly disclose the network that is a packet-based network.

DuVal clearly disclosed such a packet-based network and such a packet-based network address *(see Fig.1: block 16 (packet-based network)).*

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify such a packet-based network of DuVal to the network as taught by Dunn et al. in order to provide an anonymous or analog telephone communication system over a packet-based network (VoIP), **the motivation being** to utilize both digitally encoded voice and data over packet-based networks.

l) **In Regarding to Claim 68**: **Dunn et al. further disclosed** the method further comprising:

processing the received media signal in order to generate a speech indication signal based upon the received media signal *(see col.5 lines 22-25: The signals are processed by DSP 36)*; and

outputting the speech indication signal to the conference bridge *(see Fig.4: connections between Parties A-D and conference Bridge 14)*.

It would have been obvious to combine **Dunn et al.** and **DuVal** for the same reason as in claim 67.

15. **Claims 42, 43, 45-48, 55, 56 and 58-61** are rejected under 35 U.S.C. 103(a) as being unpatentable over **Dunn et al.** (US Patent No. **5,991,385**) in view of **DuVal** (US Patent No. **5,818,836**) as applied to claims 38-41 above, and further in view of **Julstrom** (US Patent No. **4,685,425**).

a) **Regarding to Claims 42 and 43**: **Dunn et al.** and **DuVal disclosed** all aspects of the claims 42 and 43 as set forth in the claims 38 and 41.

Both **Dunn et al.** and **DuVal failed to explicitly teach** the speech detection unit operates to determine an energy level for the received media signal and compare the determined energy level with a speech indication energy threshold (**claim 42**); and

the speech detection unit operates to determine a pitch value for the received media signal and compare the determined pitch value with a speech indication pitch threshold (**claim 43**).

However, **Julstrom disclosed** such a detection unit *(see Fig.6: Box 85 noise adapting threshold (energy threshold), box 89 comparator (compare); 56 MAX bus and see abstract: loudest microphone signal (pitch value))*.

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify such a speech detection unit of Julstrom to the DSP unit as taught by Dunn et al. in order to detect suitable speech signals in a teleconference system, **the motivation being** to make Dunn et al. more efficient.

b) **In Regarding to Claim 45**: **Dunn et al. further disclosed** the speech detection unit operates to:

determine a speech parameter corresponding to the received media signal *(col.5 lines 51-56: DSP code measures individual inputs, and the measurement of volume and power density or different frequencies of the inputs (speech parameter))*; and

include the speech parameter within the speech indication signal *(see col.5 lines 20-30: re-create the audio and output array).*

It would have been obvious to combine **Dunn et al.** and **DuVal** for the same reason as in claim 38.

c) **In Regarding to Claim 46**: **Dunn et al.** and **DuVal disclosed** all aspects of the claim 46 as set forth in the claims 38 and 45.

Both **Dunn et al.** and **DuVal failed to explicitly teach** the speech detection unit determines an energy level corresponding to the received media signal.

However, **Julstrom disclosed** such a detection unit *(see Fig.6: Box 85 noise adapting threshold (energy level), box 89 comparator (determine)).*

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify such a speech detection unit of Julstrom to the DSP unit as taught by Dunn et al. in order to detect suitable speech signals in a teleconference system, **the motivation being** to make Dunn et al. more efficient.

d) **In Regarding to Claim 47**: **Dunn et al.** and **DuVal disclosed** all aspects of the claim 47 as set forth in the claims 38 and 45.

Both **Dunn et al.** and **DuVal failed to explicitly teach** the speech detection unit determines a pitch value corresponding to the received media signal.

However, **Julstrom disclosed** such a detection unit *(see Fig.6: 56 MAX bus and see abstract: loudest microphone signal (pitch value))*.

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify such a speech detection unit of Julstrom to the DSP unit as taught by Dunn et al. in order to detect suitable speech signals in a teleconference system, **the motivation being** to make Dunn et al. more efficient.

e) **In Regarding to Claim 48**: **Dunn et al.** and **DuVal disclosed** all aspects of the claim 48 as set forth in the claims 38 and 45.

Both **Dunn et al.** and **DuVal failed to explicitly teach** the output unit further operates to compress the received media signal prior to outputting the compressed media signal to the conference bridge; and the speech detection unit determines the number of bytes of the compressed media signal. **However, Official** notice is taken that compress the received media signal prior to outputting the compressed media signal to the conference bridge and the speech detection unit determines the number of bytes of the compressed media signal is well known and accepted as standard in the subject matter area of the invention.

It would have been obvious to include in Dunn et al. and DuVal any old and well known of such operations of the output unit, since it is old and well known in the

environment of the invention and **would make Dunn et al.** and **DuVal** more efficient.

f) **Regarding to Claims 55 and 56**: **Dunn et al.** and **DuVal disclosed** all aspects of the claims 55 and 56 as set forth in the claims 49, 53 and 54.

Both **Dunn et al.** and **DuVal failed to explicitly teach** the speech detection unit operates to determine an energy level for the received media signal and compare the determined energy level with a speech indication energy threshold (**claim 55**); and

the speech detection unit operates to determine a pitch value for the received media signal and compare the determined pitch value with a speech indication pitch threshold (**claim 56**).

However, **Julstrom disclosed** such a detection unit *(see Fig.6: Box 85 noise adapting threshold (energy threshold), box 89 comparator (compare); 56 MAX bus and see abstract: loudest microphone signal (pitch value)).*

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify such a speech detection unit of Julstrom to the DSP unit as taught by Dunn et al. in order to detect suitable speech signals in a teleconference system, **the motivation being** to make Dunn et al. more efficient.

g) **In Regarding to Claim 58**: **Dunn et al. further disclosed** the speech detection unit operates to:

determine a speech parameter corresponding to the received media signal *(col.5 lines 51-56: DSP code measures individual inputs, and the measurement of volume and power density or different frequencies of the inputs (speech parameter))*; and

include the speech parameter within the speech indication signal *(see col.5 lines 20-30: re-create the audio and output array).*

It would have been obvious to combine **Dunn et al.** and **DuVal** for the same reason as in claim 49.

h) **In Regarding to Claim 59**: **Dunn et al.** and **DuVal disclosed** all aspects of the claim 59 as set forth in the claims 49, 53 and 58.

Both **Dunn et al.** and **DuVal failed to explicitly teach** the speech detection unit determines an energy level corresponding to the received media signal.

However, **Julstrom disclosed** such a detection unit *(see Fig.6: Box 85 noise adapting threshold (energy level), box 89 comparator (determine)).*

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify such a speech detection unit of Julstrom to the DSP unit as taught by Dunn et al. in order to detect suitable speech signals in a teleconference system, **the motivation being** to make Dunn et al. more efficient.

i) **In Regarding to Claim 60**: **Dunn et al.** and **DuVal disclosed** all aspects of the claim 60 as set forth in the claims 49, 53 and 58.

Both **Dunn et al.** and **DuVal failed to explicitly teach** the speech detection unit determines a pitch value corresponding to the received media signal.

However, **Julstrom disclosed** such a detection unit *(see Fig.6: 56 MAX bus and see abstract: loudest microphone signal (pitch value)).*

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify such a speech detection unit of Julstrom to the DSP unit

as taught by Dunn et al. in order to detect suitable speech signals in a teleconference system, **the motivation being** to make Dunn et al. more efficient.

j) **In Regarding to Claim 61**: **Dunn et al.** and **DuVal disclosed** all aspects of the claim 61 as set forth in the claims 49, 53 and 58.

Both **Dunn et al.** and **DuVal failed to explicitly teach** the output unit further operates to compress the received media signal prior to outputting the compressed media signal to the conference bridge; and the speech detection unit determines the number of bytes of the compressed media signal. **However, Official** notice is taken that compress the received media signal prior to outputting the compressed media signal to the conference bridge and the speech detection unit determines the number of bytes of the compressed media signal is well known and accepted as standard in the subject matter area of the invention.

It would have been obvious to include in Dunn et al. and DuVal any old and well known of such operations of the output unit, since it is old and well known in the environment of the invention and **would make Dunn et al.** and **DuVal** more efficient.

16. **Claims 44 and 57** are rejected under 35 U.S.C. 103(a) as being unpatentable over **Dunn et al.** (US Patent No. **5,991,385**) in view of **DuVal** (US Patent No. **5,818,836**) as applied to claims 38-41 above, and further in view of **Su et al.** (US Patent No. **6,463,414**).

a) Regarding to Claim 44: Both Dunn et al. and **DuVal disclosed** all aspects of the claim 44 as set forth in the claims 38 and 41.

Dunn et al. and **DuVal explicitly failed to disclose** the output unit further operates to compress the received media signal prior to outputting the media signal to the conference bridge; and wherein to determine if the received media signal contains speech, the speech detection unit operates to determine if the number of bytes of the compressed media signal indicates that the received media signal contains speech.

However, Su et al. clearly disclosed such a compressed media signal and such a number of bytes of the compressed media signal *(see col.4 lines 8-19: speech coding/compression techniques; and see G.723.1; the G.723.1 is a VoIP standard well known in the art (hence, number of bytes in a voice data packet)).*

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify such a compressed media signal and such a number of bytes of the compressed media signal of Su et al. to the voice over a packet-switching network as taught by DuVal in order to provide digitally encoded voice that can be transmitted over a packet-switched network, **the motivation being** to make DuVal compatible in different protocol networks.

b) Regarding to Claim 57: Both Dunn et al. and **DuVal disclosed** all aspects of the claim 57 as set forth in the claims 49, 53 and 54.

Dunn et al. and **DuVal explicitly failed to disclose** the output unit further operates to compress the received media signal prior to outputting the media signal to the conference bridge; and wherein to determine if the received media signal contains speech, the speech detection unit operates to determine if the number of bytes of the compressed media signal indicates that the received media signal contains speech.

However, Su et al. clearly disclosed such a compressed media signal and such a number of bytes of the compressed media signal *(see col.4 lines 8-19: speech coding/compression techniques; and see G.723.1; the G.723.1 is a VoIP standard well known in the art (hence, number of bytes in a voice data packet)).*

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify such a compressed media signal and such a number of bytes of the compressed media signal of Su et al. to the voice over a packet-switching network as taught by DuVal in order to provide digitally encoded voice that can be transmitted over a packet-switched network, **the motivation being** to make DuVal compatible in different protocol networks.

Claim Rejections - 35 USC § 102

17. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (b) the invention was patented or described in a printed publication in this or a foreign country or in public use or on sale in this country, more than one year prior to the date of application for patent in the United States.

18. **Claims 64-65** are rejected under 35 U.S.C. 102(b) as being anticipated by **Strawczynski et al.** (US Patent No. **4,920,565**).

a) **In Regarding to Claim 64**: **Strawczynski et al. disclosed** a method for controlling a media conference including at least two sources of media signals, the method comprising:

selecting a set of the sources of media signals within the media conference as talkers *(see Figs.1 and 6, and col.6 lines 16-25: in which, the role of the conference control unit 80 includes the demultiplexing (selecting) of each input channel into two logical channels and redistribution and multiplexing of these among the participants within the media conference)*; and

instructing the sources within the media conference selected as talkers to output their media signals directly to other sources within the media conference *(see Fig.2: the interconnections between telephones A-D and the connection between switching network 12 and the conference control unit 11 would be considered as that the conference control unit 11 instructs a number of the sources A-D within the media conference to output their media signals directly to other sources; and see col.3 lines 41-46: no processing of speech by the conference unit or by the network is needed (hence the talkers output their media signals directly to other sources within the media conference))* .

b) **In Regarding to Claim 65**: **Strawczynski et al. further disclosed** the selecting a set of the sources of media signals within the media conference as talkers comprises:

receiving media signals from the sources within the media conference *(see Figs.6 and 4: based on the connection between telephones A-D (sources) and the conference control unit 80 via switches A-D in Fig.6 and the outputs 59 and 62 shown in Fig.4, that would be considered the conference control unit 80 would receive media signals from the sources A-D within the media conference)*;

determining at least one speech parameter corresponding to each of the received media signals *(see col.6 lines 30-38: bit patterns on two channels (speech parameter))*; and

selecting a set of the sources within the media conference as talkers based on the determined speech parameters *(see col.6 lines 45-68: The conference control unit 11 is responsible for administrating the connection between the parties and reconnect the remaining parties (hence, selecting a set of the sources within the media conference as talkers based on the determined speech parameters).*

Claim Rejections - 35 USC § 102

19. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (e) the invention was described in (1) an application for patent, published under section 122(b), by another filed in the United States before the invention by the applicant for patent or (2) a patent granted on an application for patent by another filed in the United States before the invention by the applicant for patent, except that an international application filed under the treaty defined in section 351(a) shall have the effects for purposes of this subsection of an application filed in the United States only if the international application designated the United States and was published under Article 21(2) of such treaty in the English language.

20. **Claims 64-66** are rejected under 35 U.S.C. 102(e) as being anticipated by **Boyle et al.** (US Patent No. **6,606,305**).

a) **In Regarding to Claim 64**: **Boyle et al. disclosed** a method for controlling a media conference including at least two sources of media signals *(see Fig.1)*, the method comprising:

selecting a set of the sources of media signals within the media conference as talkers *(see col.5 lines 41-48: the selected subscriber group)*; and

instructing the sources within the media conference selected as talkers to output their media signals directly to other sources within the media conference *(see col.13 lines 50-65: the conference bridge including instructions to receive; and see col.9 lines 27-57: The database 320 and conference bridge 330 are each connected to lines 345, such as SS7 or ISUP trunks (hence the bridge 330 in Fig.3 is used for controlling and connection set up only; therefore, the media signals of the talker would be outputted directly to other sources within the media conference via PSTN and Mobile switching center))* .

b) **In Regarding to Claim 65**: **Boyle et al. further disclosed** the selecting a set of the sources of media signals within the media conference as talkers comprises:

receiving media signals from the sources within the media conference *(see col.13 lines 50-65: the conference bridge including instructions to receive the first coming call leg to determine a plurality of directory numbers associated with subscriber group (hence, receiving media signals from the sources within the media conference); also, see Fig.2A: step 200)*;

determining at least one speech parameter corresponding to each of the received media signals *(see Fig.2A: step 205)*; and

selecting a set of the sources within the media conference as talkers based on the determined speech parameters *(see Fig.2A: steps 210-225)*.

c) **In Regarding to Claim 66**: **Boyle et al. further disclosed** the selecting a set of the sources of media signals within the media conference as talkers comprises:

receiving speech indication signals from at least one of the sources within the media conference *(see col.9 lines 27-42: the switch 130 or MSC 310 has received an*

indication that the incoming call leg is for a voice dispatch conferencing or broadcast feature, requiring routing to the conference bridge 130 or 330); and

selecting a set of the sources within the media conference as talkers based on the received speech indication signals *(see col.9 lines 27-42: and the switch 110 or MSC 310 may then utilize the subscriber group number or the predefined pseudo-random number mapped to the subscriber number as destination digits (hence, selecting a set of sources as talkers based on the receive indication signals)).*

Conclusion

21. Any inquiry concerning this communication or earlier communications from the examiner should be directed to **Anthony T Ton** whose telephone number is 703-305-8956. The examiner can normally be reached on M-F: 8:00 am - 4:30 pm.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Douglas W Olms can be reached on 703-305-4703. The fax phone number for the organization where this application or proceeding is assigned is 703-872-9306. Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

ATT 3/31/2004

KENNETH VANDERPUYE
PRIMARY EXAMINER